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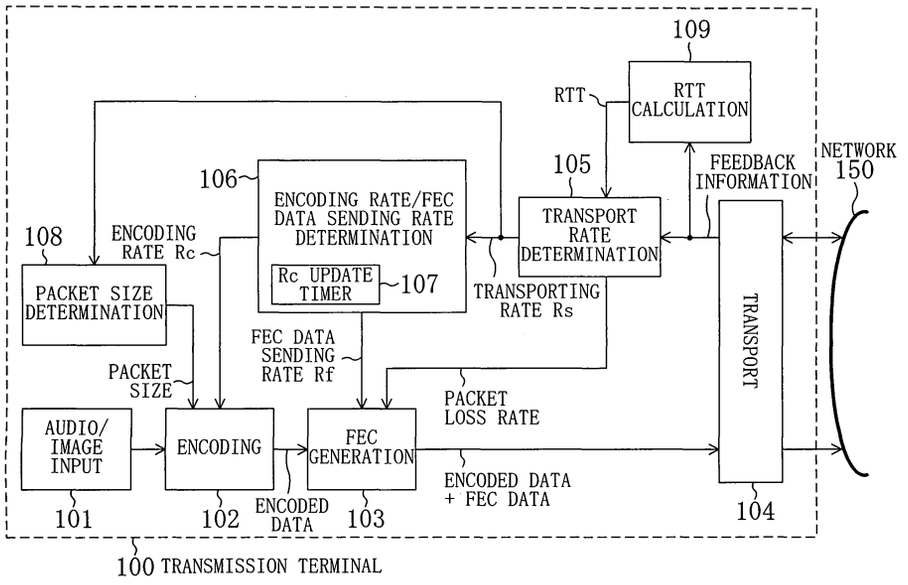
(54) Method, transmitter and receiver for adapting the encoding rate to a varying transport rate

(57) When a transport rate **Rs** is updated, in the case where the updated transport rate **Rs** is within a variable range (**Rc** < **Rs** < **U** x **Rc**, and **U** > 1) defined by a current encoding rate **Rc**, FEC (Forward Error Correction) data sending rate **Rf** is updated (**Rf** = **Rs - Rc**) without updating the current encoding rate **Rc**. On the other hand, in the case where an Rc update timer set to three seconds, for example, is timed out, the encoding rate **Rc** is updated using a minimum value **Rs_min** of the transport rate up to that time (**Rc** = **V** x **Rs_min**, and 0 < **V** < 1). Even when the transport rate **Rs** greatly varies to be out of the variable range, the encoding rate **Rc** is updated in the same manner.

FIG. 1



EP 1 372 304 A2

Description

BACKGROUND OF THE INVENTION

[0001] The present invention relates to a method for transporting audio/video data transmitted from a camera or a microphone, that is, media data with high quality, a transmitter and a receiver therefor.

[0002] Various connection modes are used over an IP (Internet Protocol) network, such as an intranet or the Internet. Depending on the connection mode, the usable bands of the network in terms of bit rate vary in a range of from several kilobits per second (kbps) to several hundreds of megabits per second (Mbps). In addition, the influence of other flows (transmission data) causes the usable band to vary in time. As such, transport rate control is necessary to perform media data transport across a network of the type described above. The transport rate control adjusts the transport rate in accordance with the usable transport bands.

[0003] Known protocols employed to transport media data across the Internet in real time include RTP (Realtime Transport Protocol)/RTCP (RTP Control Protocol) (H. Schulzrinne et al., "RTP: A Transport Protocol for Real-Time Applications", RFC 1889, Internet Engineering Taskforce, Jan. 1996). According to the RTP/RTCP, a reception terminal transmits feedback information regarding, for example, the packet loss rate, propagation delay time, and jitter, to a transmission terminal, and the transport rate is determined on the basis of the feedback information. The transmission interval (notification interval) for the feedback information from the reception terminal is a fixed value, for example, five seconds.

[0004] Known methods of transport rate control using feedback information as described above include a method called TFRC (TCP Friendly Rate Control) (M. Handley et. al., "TCP Friendly Rate Control (TFRC): Protocol Specification", RFC 3448, Internet Engineering Taskforce, Jan. 2003). According to the TFRC, the transport rate is controlled to increase until a packet loss event occurs, and the transport rate is controlled to decrease when the packet loss event has occurred. Thereby, TFRC enables update to be smoothly achieved for producing a transport rate suitable for intended media data transmission. The transport rate update interval is 1 x RTT (Round Trip Time), ranging from several tens of milliseconds (ms) to several hundreds of milliseconds (ms), for example.

[0005] Another known method is a method described in Request for Comments (RFC) 2733 (J. Rosenberg et al., "An RTP Payload Format for Generic Forward Error Correction", Internet Engineering Taskforce, Dec. 1999), which is a technique of imparting an error correction capability to RTP. More specifically, the RFC 2733 technique adds FEC data (FEC: forward error correction), which is redundant data for forward error correction, to data which is to be transmitted.

[0006] In the event of transmitting encoded data based on media data, when its transport rate is updated following the TFRC at intervals ranging from several tens of milliseconds (ms) to several hundreds of milliseconds (ms), and the encoding rate is updated every time the transport rate is updated, the encoding rate frequently varies. As such, for example, quality of images to be transmitted frequently varies, resulting in deteriorating view image quality.

[0007] In addition, when the notification interval for feedback information from a reception terminal to a transmission terminal is set short, an advantage is exhibited in that a usable transmission band can quickly be accessed, and the transmission band can be efficiently used. On the other hand, the update frequency for the transport rate in the transmission terminal is rendered high, and quality of media data is frequently varied to an extent that the media transport cannot be implemented maintaining stable quality. In addition, the degree of fairness in throughput with respect other flow is reduced. Conversely, with the notification interval for the feedback information is set long, the update frequency for the transport rate is reduced. In this case, while the transport rate is stabilized, variations in the transmission band cannot be followed, thereby disabling efficient use of the transmission band. Further, in the case where the notification interval is set long, when the usable transmission band is narrow, packets are lost in a burst-wise manner, and media data transmission quality is extremely deteriorated.

[0008] Moreover, conventionally, since the data packet size is uniform, when the transport rate is reduced, the number of the packets is reduced. Generally, in a reception terminal, a packet loss rate **L** is calculated by the following equation:

$$L = 1 - (\text{number of data packages arrived at a reception terminal within a time } T)/(\text{number of data packages transmitted from a transmission terminal within the time } T)$$

[0009] Ordinarily, the time **T** is a time interval from the instant of transmission of previous feedback information to the instant of transmission of current feedback information. The equation describes that as the number of packets to be transported from the transmission terminal decreases, the packet loss of one packet causes the packet loss rate to be greatly vary. For example, assume a state where the transport rate significantly decreases, and no more than one packet can be transported within the time **T**. In this state, when the packet is not lost, the packet loss rate is 0%;

whereas, when the packet is lost, the packet loss rate is 100%. In such the state, for the reason that an accurate network congestion state cannot secured for the reason that the packet loss rate cannot finely be observed.

[0010] Further, in a method represented by TFRC and placing primary importance on the fairness with respect to the TCP, The RTT is included in a denominator of a calculation expression that determines the transport rate. The RTT is a time required for the round trip of a packet between a transmission terminal and a reception terminal; as such it can take a value 1 ms or less when the distance between the transmission terminal and the reception terminal is short. In this state, the RTT measurement result is influenced by various delay fluctuations irrelevant to the network congestion state. For example, the delay fluctuations include a delay fluctuation in processing of an operating system that performs processing for packet transmission from the transmission terminal, and a delay fluctuation in transfer processing of a router that performs packet transfer processing.

SUMMARY OF THE INVENTION

[0011] An object of the present invention is to implement high-quality media data transport under an environment in which a usable band greatly varies.

[0012] In order to achieve the object, the present invention is directed to suppression of frequent update events for encoding rates, adjustment of a notification interval for feedback information, change of the packet size, and addition of an offset to an RTT. Consequently, high-quality media data transport under an environment in which a usable band greatly varies can be implemented. Therefore, the present invention can be applied to various applications, such as a VoIP (Voice Over Internet Protocol) utilized for Internet telephones, a TV telephone, and monitoring/broadcasting utilizing a network.

[0013] More specifically, in transport of encoded data based on media data, the present invention is directed to update a transport rate in accordance with a transport state of the network and to update an encoding rate at a frequency lower than an update frequency for the transport rate. For example, when a timer set to a timeout period longer than an update interval for the transport rate is timed out or when the transport rate is updated to a value which is out of a predetermined range, the transport rate is updated in accordance with a minimum value of the transport rate within a time. In the case of adding redundant data for error correction to the encoded data, a difference between the transport rate and the encoding rate may be allocated to a sending rate for redundant data.

[0014] In addition, the present invention is directed to notify of feedback information indicative of a transport state of the network from a reception terminal at a variable interval in accordance with the transport state of the network and to update a transport rate for the media data in accordance with the notified feedback information. For example, a notification interval for the feedback information is shortened when a packet loss rate is high, and the notification interval for the feedback information is lengthened when the packet loss rate is low. Preferably, the notification interval of the feedback information is limited in such a manner that it is not shorter than a predetermined minimum interval.

[0015] Further, in transport of media data, the present invention is directed to change the data packet size to a small value when the transport rate is lowered and a predetermined number of packets is determined not to be transmittable within a packet-loss observation period. In order to precisely gain a congestion state of the network, the number of data packages to be sent from a transmission terminal within a predetermined period may be set to a predetermined number or more. It can be considered that the data packet size is preset small and the number of packets to be sent is increased. However, since a header is provided to the packet, a problem arises in that the overhead of the headers is increased when the number of packets is increased. As such, according to the present invention, an adoptive process is performed in such a manner that when the transport rate is high, the packet size is set to be large, and when the transport rate is lowered, the packet size is set to be small, whereby the observation accuracy for the packet loss rates is improved. Consequently, the packet loss rates can be finely observed, and accurate transport rates can be determined.

[0016] Further, the present invention is directed to add an offset to an RTT when the RTT is smaller than a predetermined value. Consequently, when the value of the RTT is excessively small, the transport rate can be prevented from being greatly influenced by fluctuation of a delay irrelevant to the congestion of the network.

BRIEF DESCRIPTION OF THE DRAWINGS

[0017]

Fig. **1** is a block diagram showing an internal configuration example of a media transmitter (transmission terminal) according to the present invention.

Fig. **2** is a block diagram showing an internal configuration example of a media receiver (reception terminal) according to the present invention.

Fig. **3** is a flowchart showing an initial operation of an encoding rate/FEC data sending rate determination section

shown in Fig. **1**.

Fig. **4** is a flowchart showing an interrupt operation of the aforementioned determination section.

Fig. **5** is a flowchart showing main operations of a feedback information generation section and notification interval control section shown in Fig. **2**.

Fig. **6** is a timing chart showing variation examples of a transport rate and an encoding rate in media data transmissions by the devices shown in Figs. **1** and **2**.

Fig. **7** is a flowchart showing an operation of a packet size determination section shown in Fig. **1**.

Fig. **8** is a flowchart showing an operation of an RTT calculation section shown in Fig. **1**.

Fig. **9** is a graph for describing RTT offset addition implemented by the operation shown in Fig. **8**.

DETAILED DESCRIPTION OF THE INVENTION

[0018] Hereinafter, embodiments of a method for transporting media, a transmitter and a receiver therefor according to the present invention with reference to the drawings.

[0019] Figs. **1** and **2** show a configuration example of a media transport system for transporting encoded data based on media data. Specifically, Fig. **1** shows an internal configuration example of a media transmitter (transmission terminal) **100**, and Fig. **2** shows an internal configuration example of a media receiver (reception terminal) **200**. The transmission terminal **100** and the reception terminal **200** are connected together a network **150**. The network **150** may be a wired network, a wireless network, or a network including alternately connected wired and wireless networks. The system employs, for example, the realtime transport protocol/realtime transport control protocol (RTP/RTCP) as transport protocols between the transmission terminal **100** and the reception terminal **200**.

[0020] The transmission terminal **100** of Fig. **1** has a configuration including an audio/image input section **101**, an encoding section **102**, an FEC generation section **103**, a transport section **104**, a transport rate determination section **105**, an encoding rate/FEC data sending rate determination section **106**, a packet size determination section **108**, and an RTT calculation section **109**. The audio/image input section **101** is, for example, a camera or a microphone for inputting audio/video data (media data) to be encoded. The encoding section **102** generates encoded data from media data in accordance with a specified encoding rate **Rc**. For example, standardized compressing/encoding schemes, such as MPEG (Moving Picture Coding Experts Group) 1/2/4 and AMR (Audio/Modem Riser), may be employed. The encoding section **102** has the function of outputting data with a packet size determined by the packet size determination section **108**. The FEC generation section **103** generates FEC data, which is redundant data for error correction, from encoded data in accordance with a specified FEC data sending rate **Rf**. The transport section **104** transmits the encoded data and the FEC data to the network **150** and transmits and receives control information to enable feedback information (such as packet loss rates, propagation delay times, and jitters) indicative of the transport state of the network **150** to be secured from the reception terminal **200**. The transport rate determination section **105** updates a transport rate **Rs** in accordance with the feedback information and an RTT calculated by the RTT calculation section **109**, and notifies the FEC generation section **103** of a packet loss rate. For example, a TFRC is employed as an algorithm for determining the transport rate **Rs**. The encoding rate/FEC data sending rate determination section **106** updates the encoding rate **Rc** at an update frequency lower than an update frequency for the transport rate **Rs** by using an **Rc** update timer 107, and allocates a difference between the transport rate **Rs** and the encoding rate **Rc** to the FEC data sending rate **Rf**. The packet size determination section **108** determines a packet size in accordance with the transport rate **Rs** notified from the transport rate determination section **105**, and notifies the encoding section **102** of the determined result. The RTT calculation section **109** receives the feedback information from the transport section **104**, and calculates a round trip propagation delay time (RTT), used for calculating the transport rate **Rs**, from the feedback information.

[0021] The reception terminal **200** of Fig. **2** has a configuration including a transport section **201,** a packet-loss recovery section **202**, a decoding section **203**, a reproduction section **204**, a feedback information generation section **205**, and a notification interval control section **206**. The transport section **201** receives the encoded data and the FEC data transported from the transmission terminal **100** via the network **150**, and transmits and receives control information to enable the generation of feedback information indicative of a transport state of the network **150**. The packet-loss recovery section **202** uses the FEC data to recover encoded data in a packet lost in the network **150**. The decoding section **203** decodes the encoded data to secure media data. The reproduction section **204** is, for example, a display or a speaker for reproducing the decoded result. The feedback information generation section **205** generates feedback information indicative of the transport state of the network **150**. the transport section **201** notifies the transmission terminal **100** of the generated feedback information via the network **150**. The notification interval control section **206** controls a notification interval **Tf** of the feedback information in accordance with the transport state of the network **150**.

[0022] Fig. **3** shows an initial operation of the encoding rate/FEC data sending rate determination section **106** shown in Fig. **1**. Upon activation of the transmission terminal **100**, the encoding rate/FEC data sending rate determination section **106** first acquires values of constants **U** and **V**. The constant **U** is a value (**U** > 1) for defining a narrow variable range of the transport rate **Rs**. The constant **V** is a value (0 < **V** < 1) used to control the encoding rate **Rc** not to exceed

the transport rate **Rs**. Next, the encoding rate/FEC data sending rate determination section **106** acquires a timeout period To (longer than 1 × RTT: three seconds, for example) and an initial value of the transport rate **Rs.** Subsequently, an initial value of the encoding rate **Rc** is set to **V** × **Rs**, and an initial value of the FEC data sending rate **Rf** is set to **Rs - Rc**. Then, a maximum transport rate **Rs_max** is set to a variable **Rs_min**, and the Rc update timer **107** is started, whereby the encoding rate/FEC data sending rate determination section **106** turns into a WAIT state (standby state). The **Rs_min** is a variable for acquiring the minimum value of the transport rate **Rs** within a time.

[0023] Fig. **4** shows an interrupt operation of the encoding rate/FEC data sending rate determination section **106** shown in Fig. **1**. The instance when an interrupt occurs in the encoding rate/FEC data sending rate determination section **106** and causes the encoding rate/FEC data sending rate determination section **106** to leave the WAIT state occurs when the transport rate **Rs** is updated by the transport rate determination section **105** or when the **Rc** update timer **107** is timed out. The transport rate **Rs** is assumed to be updated by the transport rate determination section **105** following the TFRC every time feedback information is received from the reception terminal **200**.

[0024] When an updated transport rate **Rs** is notified to the encoding rate/FEC data sending rate determination section **106** (step **301**), it is checked whether the notified transport rate **Rs** is within a variable range (**Rc** < **Rs** < **U** × **Rc**) defined by the current encoding rate **Rc** (step **302**). When the transport rate **Rs** is within the variable range, the encoding rate/FEC data sending rate determination section **106** updates an FEC data sending rate **Rf** in accordance with the notified transport rate Rs without updating the current encoding rate **Rc** (**Rf** = **Rs - Rc**: step **303**). When the notified transport rate **Rs** is a minimum transport rate during a period from the activation of the **Rc** update timer **107**, the encoding rate/FEC data sending rate determination section **106** preserves it into the variable **Rs_min** (step **304**), and returns to the WAIT state.

[0025] When timeout of the Rc update timer **107** occurs (step **305**), the encoding rate/FEC data sending rate determination section **106** updates the encoding rate **Rc** (**Rc** = **V** × **Rs_min**) using the value of the variable **Rs_min** in that state; updates the FEC data sending rate Rf in accordance with the current transport rate **Rs** and the updated encoding rate **Rc (Rf = Rs - Rc**); reactivates the **Rc** update timer **107**; and sets the maximum transport rate **Rs_max** again to the variable **Rs_min** (step **306**); and then returns to the WAIT state.

[0026] When the transport rate **Rs** is varied to be out of the variable range (**Rc < Rs < U** × **Rc**), the encoding rate/FEC data sending rate determination section **106** sets the notified transport rate **Rs** to the variable **Rs_min** (step **307**); performs the operation in step **306** such as update of the encoding rate **Rc**; and then returns to the WAIT state.

[0027] Thus, the algorithm described above enables the update frequency for the encoding rate **Rc** to be suppressed and enables an appropriate FEC data sending rate **Rf** to be secured.

[0028] Next, an operation example of the FEC generation section **103** of Fig. **1** will be described. FEC data generation uses, for example, the XOR (exclusive-OR) operation specified in the RFC-2733. In brief, with fixed error tolerability being premised, FEC data is generated to enable the loss of, for example, two continuing packets, to be recovered. However, the error correction capability may be modified after detecting the number of continuously lost packets. When the amount of generated FEC data is smaller than the FEC data sending rate **Rf** determined according to the algorithm shown in Figs. **3** and **4**, the FEC data are all sent out. When the amount of FEC data is greater than the determined encoding rate **Rc,** the amount of the FEC data is adjusted. When the amount of FEC data to be sent out is too great, the amount of the FEC data may be reduced by lowering the error correction capability. In addition to the scheme using the XOR operation, a different error correction scheme, such as a Reed-Solomon or parity scheme, may be used.

[0029] Alternatively, the operation may be arranged such that a plurality of error correction schemes are employed to be dynamically switchable corresponding to the FEC data sending rate **Rf** and the packet loss rate. For example, the operation is arranged such that a high error tolerability of a combined pattern of Reed-Solomon and interleave schemes is selected when the packet loss rate is high, and a low error tolerability of a pattern such as a parity-scheme pattern is selected when the packet loss rate is low.

[0030] When the FEC-data sending rate **Rf** is higher than a threshold **D2** (fixed value) during a predetermined period, the parity scheme with low error tolerability is used while the process amount of error correction operation is small. In contrast, when the FEC data sending rate **Rf** is lower than or equal to the threshold **D2** during a predetermined period, the Reed-Solomon scheme with high error tolerability is used while the process amount of error correction operation is large. According to this arrangement, when the send-out amount of FEC data is large, the parity scheme is used to enable the processing load of the reception terminal **200** to be reduced. Concurrently, when the send-out amount of FEC data is small, the Reed-Solomon scheme is used to enable the error tolerability to be maintained high. Reducing the processing load of the reception terminal **200** provides advantages. For example, an increased amount of video data can be received from the transmission terminal **100**; and in addition, since the processing delay is reduced, media-data reproduction can be implemented with a reduced delay.

[0031] In addition, the FEC generation section **103** may be arranged such that, for example, data to which error tolerabilities are added is selected, and the number of items of the data is determined in accordance with the packet loss rate. When the packet loss rate is high, the error tolerability is selectively imparted to data important for decoding, such as intraframes and video headers for video data and voice portions for audio data.

[0032] The above will be described below in more detail. For MPEG-2 video encoded data, when the packet loss rate is low, a scheme of adding FEC data to the entirety of the video encoded data (the scheme will be referred to as a scheme X). When the packet loss rate is high, a scheme is employed in which while FEC data is added only to I (Intra) and P (Predictive) frames, no FEC data is added to B (Bidirectionally predictive) frames (the scheme will be referred to as a scheme Y). Since the amounts of FEC data are identical in the schemes X and Y, the error tolerability imparted to the I and P frames in the scheme X is lower than that in the scheme Y.

[0033] Thus, according to the above arrangement in which the schemes X and Y are switched corresponding to the packet loss rate, the number of frames that can be reprduced in the reception terminal **200** can be increased greater than in the case where the schemes X and Y used independently of each other. For example, when the packet loss rate is low, since lost packets can be recovered even with a low error correction capability, the number of reproduction frames is relatively larger in the scheme X. On the other hand, however, according to the scheme X, when the packet loss rate is high, the number of I and P frames to be reproduced is small. As such, P and B frames for referencing these reproduction frames cannot be reproduced, and the number of reproduction frames sharply decreases. Unlike this case, according to the scheme Y, since the high error tolerability is imparted to the I and P frames, even when the packet loss rate is high, referenced frames (I and P frames) are not lost and, consequently, the number of reproduction frames is large.

[0034] Thus, description has been given regarding the case where I, P and B frames are referred to as targets. However, the case may be such that with hierarchical encoding being employed, base layers are used instead of the I and P frames, and extended layers are used instead of the B frames.

[0035] Fig. **5** is a flowchart showing main operations of the feedback information generation section **205** and the notification interval control section **206**, which are shown in Fig. **2**. The feedback information generation section **205** measures the packet loss rate in cooperation with the transport section **201** (step **401**). When the packet loss rate is higher than a threshold **L1** (fixed value), the notification interval control section **206** reduces a notification interval **Tf** of feedback information to be shorter than the current notification interval by a fixed value **Tc1** within a range not to be smaller than a minimum value **Tfmin** (step **402**). This causes the transport rate **Rs** to quickly follow a usable transmission band, thereby enabling occurrences of packet loss events to lower. When the packet loss rate is lower than a threshold **L2** (fixed value smaller than **L1**), the notification interval control section **206** increases the notification interval **Tf** of feedback information to be longer than the current notification interval by a fixed value **Tc2** (step **403**). This enables frequent switching operations for the transport rate to decrease, and further enables fairness with the other flows to be improved.

[0036] Fig. **6** shows variation examples in the transport rate **Rs** and the encoding rate **Rc** in media data transmission by the transmission terminal **100** and reception terminal **200** shown in Figs. **1** and **2**, respectively. According to Fig. **6**, even under a state where the transport rate **Rs** is frequently updated, the operation of the encoding rate/FEC data sending rate determination section **106** reduces the update frequency for the encoding rate **Rc**. In addition, when the packet loss rate is low, the notification interval control section **206** operates to reduce unnecessary updates of the transport rate **Rs.**

[0037] Fig. **7** is a flowchart showing an operation of the packet size determination section **108** shown in Fig. **1**. According to the operation shown in Fig. **7**, when the transport rate **Rs** decreases, the packet loss rate can finely be observed in a manner that the data packet size is reduced.

[0038] First, the packet size determination section **108** receives a transport rate **Rs** determined by the transport rate determination section **105** (step **701**). Then, with the letter **S** representing the packet size in a regular mode, the operation obtains a number **N** of packets transportable at the transport rate **Rs** during an observation period **T**, as follows:

$$N = Rs \times T/S.$$

[0039] Then, the packet size determination section **108** determines as to whether the value of the number **N** of packets is smaller than a threshold Nmin (step **702**). As a result, when **N** is determined not to be smaller than **Nmin**, the regular-mode packet size **S** is employed (step **703**). When **N** is determined smaller than **Nmin**, a packet size **S'** to be employed is calculated as follows (step **704**):

$$S' = Rs \times T/Nmin.$$

[0040] Then, the packet size determination section **108** notifies the encoding section **102** of the packet size determined in step **703** or **704**, and the procedure is finished.

[0041] As described with reference to Fig. 5, according to the embodiment of Figs. **1** and **2**, since the notification interval **Tf** of the feedback information varies, the observation period **T** varies, accordingly. However, using the minimum

value **Tfmin** of the feedback information observation period enables at least the number **Nmin** of packets to be transmitted.

[0042] Fig. **8** shows an operation of the RTT calculation section **109** shown in Fig. **1**. According to Fig. **8**, an offset is added to the RTT, thereby enabling reduction to be implemented in fluctuation of the transport rate **Rs**, the fluctuation occurring in a delay irrelevant to a congestion state of the network **150**.

[0043] First, the RTT calculation section **109** receives feedback information from the reception terminal **200** (step **801**), then acquires a measured round trip propagation delay time **RTTt** from the feedback information (step **802**). Subsequently, the RTT calculation section **109** determines as to whether the **RTTt** is smaller than a predetermined threshold **RTTmin** (step **803**). When the **RTTt** is not smaller than the **RTTmin**, the **RTTt** is used as a round trip propagation delay time **RTTcalc** that is used for calculation of a transport rate **Rs** (step **804**). When the **RTTt** is smaller than the **RTTmin**, the round trip propagation delay time **RTTcalc** used for the calculation is obtained using the following equation (step **805**):

$$RTTcalc = RTTt + ofs \times (RTTmin - RTTt)/RTTmin$$

[0044] In the equation, **ofc** is a predetermined constant. Then, the RTT calculation section **109** notifies the transport rate determination section **105** of the **RTTcalc** determined in step **804** or **805** (step **806**), and then the procedure is finished.

[0045] Fig. **9** shows the relationship between the **RTTt** and the **RTTcalc** described above. In Fig. **9**, the horizontal axis indicates the **RTTt**, and the vertical axis indicates the **RTTcalc**. The relationship between the **RTTt** and the **RTTcalc** is represented by the thick solid polyline. The broken line represents a case where **RTTcalc** = **RTTt**; that is, the measured **RTT** is used as it is for the calculation. As shown in Fig. **9**, a positive addition offset is used when the **RTTt** is smaller than the **RTTmin**. As such, no case occurs in which the **RTTcalc** is excessively small even when the **RTTt** is small.

[0046] The increase in the congestion-dependent propagation delay is several hundreds of milliseconds or greater, and the processing delay of a device is on the order of several milliseconds. As such, with the value such as the above-described R**TTmin** or **ofs** being set to a value on the order of several tens of milliseconds or less, no event occurs in which the addition offset influences the propagation delay time when congestion has occurred.

[0047] The transport protocol between the transmission terminal **100** and the reception terminal **200** may be a unique protocol instead of IP. In addition, the connection is not limited to be of the one-to-one type, and may be of a one-to-N broadcast, multicast type; or it may be of an N-to-M mesh type. Media data may either be transmitted alone or be parallel transmitted in synchronization. Further, not only the standardized encoding method, but also a different unique encoding method may be used.

Claims

1. A method for transporting encoded data based on media data via a network, comprising:

a first step of updating a transport rate in accordance with a transport state of the network; and
a second step of updating an encoding rate at a frequency lower than an update frequency for the transport rate.

2. The method according to claim 1,
wherein the second step includes a step of updating the encoding rate when a timer times out, the timer being set to a timeout period longer than an update interval for the transport rate.

3. The method according to claim 1,
wherein the second step includes a step of updating the transport rate when the transport rate is updated to a value which is out of a predetermined range.

4. The method according to claim 1,
wherein the second step includes a step of updating the transport rate in accordance with a minimum value of the transport rate within a time.

5. The method according to claim 1, further comprising:

a third step of allocating a difference between the transport rate and the encoding rate to a sending rate for

redundant data used for error correction, adding the redundant data to the encoded data, and transmitting the resulting data.

6. The method according to claim 5,
wherein the third step includes a step of changing an error correction scheme in accordance with the sending rate of the redundant data.

7. The method according to claim 5,
wherein the third step includes a step of changing the error correction scheme in accordance with a packet loss rate of the network.

8. The method according to claim 7,
wherein the third step includes a step of imparting an error tolerability to all encoded data when the packet loss rate of the network is low, and imparting the error tolerability only to encoded data having a high importance when the packet loss rate of the network is high.

9. The method according to claim 1,
wherein the first step includes a step of notifying of feedback information indicative of a transport state of the network from a reception terminal.

10. The method according to claim 9, further comprising:

a step of variably controlling a notification interval for the feedback information in accordance with the transport state of the network.

11. A method for transporting media data via a network, comprising:

a first step of notifying of feedback information indicative of a transport state of the network from a reception terminal at a variable interval in accordance with the transport state of the network; and
a second step of updating a transport rate for the media data in accordance with the notified feedback information.

12. The method according to claim 11,
wherein first step includes a step of shortening a notification interval for the feedback information when a packet loss rate of the network is high, and lengthening the notification interval for the feedback information when the packet loss rate of the network is low.

13. The method according to claim 12, further comprising:

a step of limiting the notification interval of the feedback information not to be shorter than a predetermined minimum interval.

14. A media transmitter for transmitting encoded data based on media data to a network, comprising:

means for generating the encoded data from the media data;
means for generating redundant data for error correction from the encoded data;
means for updating a transport rate in accordance with a transport state of the network;
means for updating an encoding rate at a frequency lower than an update frequency for the transport rate;
means for allocating a difference between the transport rate and the encoding rate to a sending rate for the redundant data; and
means for adding the redundant data to the encoded data and transmitting the resulting data.

15. A media receiver for receiving media data from a network, comprising:

means for notifying of feedback information indicative of a transport state of the network in such a manner that a transport rate for the media data is updated; and
means for controlling a notification interval for the feedback information in accordance with the transport state of the network.

16. A method for transporting encoded data based on media data via a network, comprising:

a step of updating a transport rate in accordance with a transport state of the network;
a step of calculating the number of packets transportable within an observation period for a packet loss in the network on the basis of the updated transport rate, the length of the observation period, and a packet size of the encoded data in a regular mode;
a step of comparing the calculated number of packets with a preset threshold; and
a step of setting a packet size of the encoded data to be smaller than the packet size in the regular mode when the calculated number of packets is smaller than the threshold.

17. A media transmitter for transmitting encoded data based on media data to a network, comprising:

means for generating the encoded data from the media data;
means for updating a transport rate in accordance with a transport state of the network;
means for calculating the number of packets transportable within an observation period for a packet loss in the network on the basis of the updated transport rate, the length of the observation period, and a packet size of the encoded data in a regular mode;
means for comparing the calculated number of packets with a preset threshold;
means for setting a packet size of the encoded data to be smaller than the packet size in the regular mode when the calculated number of packets is smaller than the threshold; and
means for transmitting the encoded data.

18. A method for transmitting encoded data based on media data via a network, comprising:

a step of causing a transmission terminal to measure a round trip propagation delay time between the transmission terminal and a reception terminal;
a step of comparing the measured round trip propagation delay time to a preset threshold;
a step of adding an offset value to the measured round trip propagation delay time when the measured round trip propagation delay time is smaller than the threshold; and
a step of determining a transport rate on the basis of the round trip propagation delay time to which the offset value is added.

19. A media transmitter for transmitting encoded data based on media data to a network, comprising:

means for generating the encoded data from the media data;
means for measuring a round trip propagation delay time on the network;
means for comparing the measured round trip propagation delay time with a preset threshold;
means for adding an offset value to the measured round trip propagation delay time when the measured round trip propagation delay time is smaller than the threshold;
means for determining a transport rate on the basis of the round trip propagation delay time to which the offset value is added; and
means for transmitting the encoded data at the determined transport rate.

FIG. 1

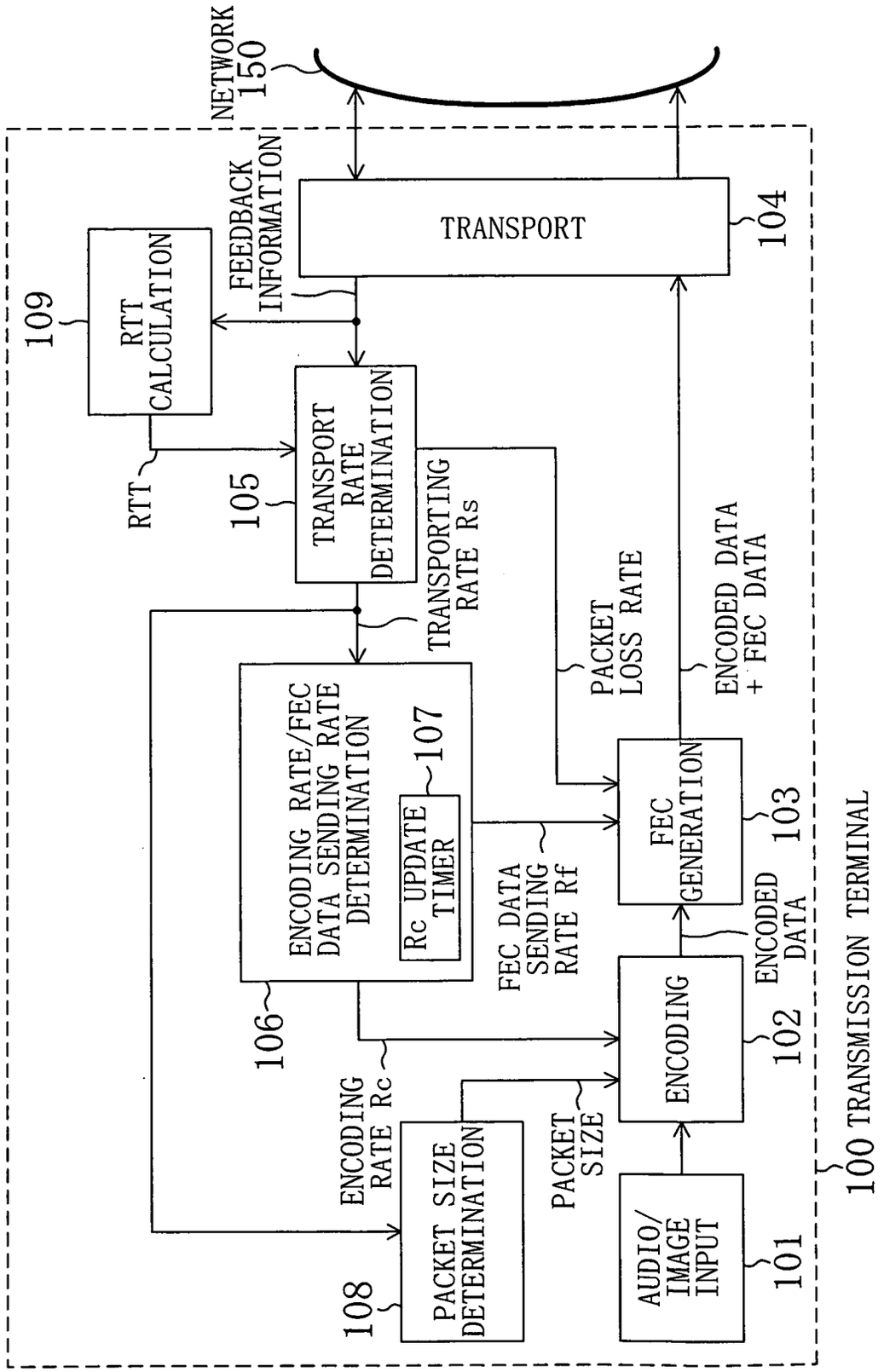

NETWORK
150
TRANSPORT
104
FEEDBACK INFORMATION
RTT CALCULATION
109
RTT
TRANSPORT RATE DETERMINATION
105
TRANSPORTING RATE Rs
PACKET LOSS RATE
ENCODED DATA + FEC DATA
ENCODING RATE/FEC DATA SENDING RATE DETERMINATION
107
Rc UPDATE TIMER
106
FEC DATA SENDING RATE Rf
FEC GENERATION
103
ENCODED DATA
ENCODING RATE Rc
PACKET SIZE
ENCODING
102
PACKET SIZE DETERMINATION
108
AUDIO/ IMAGE INPUT
101
100 TRANSMISSION TERMINAL

FIG. 2

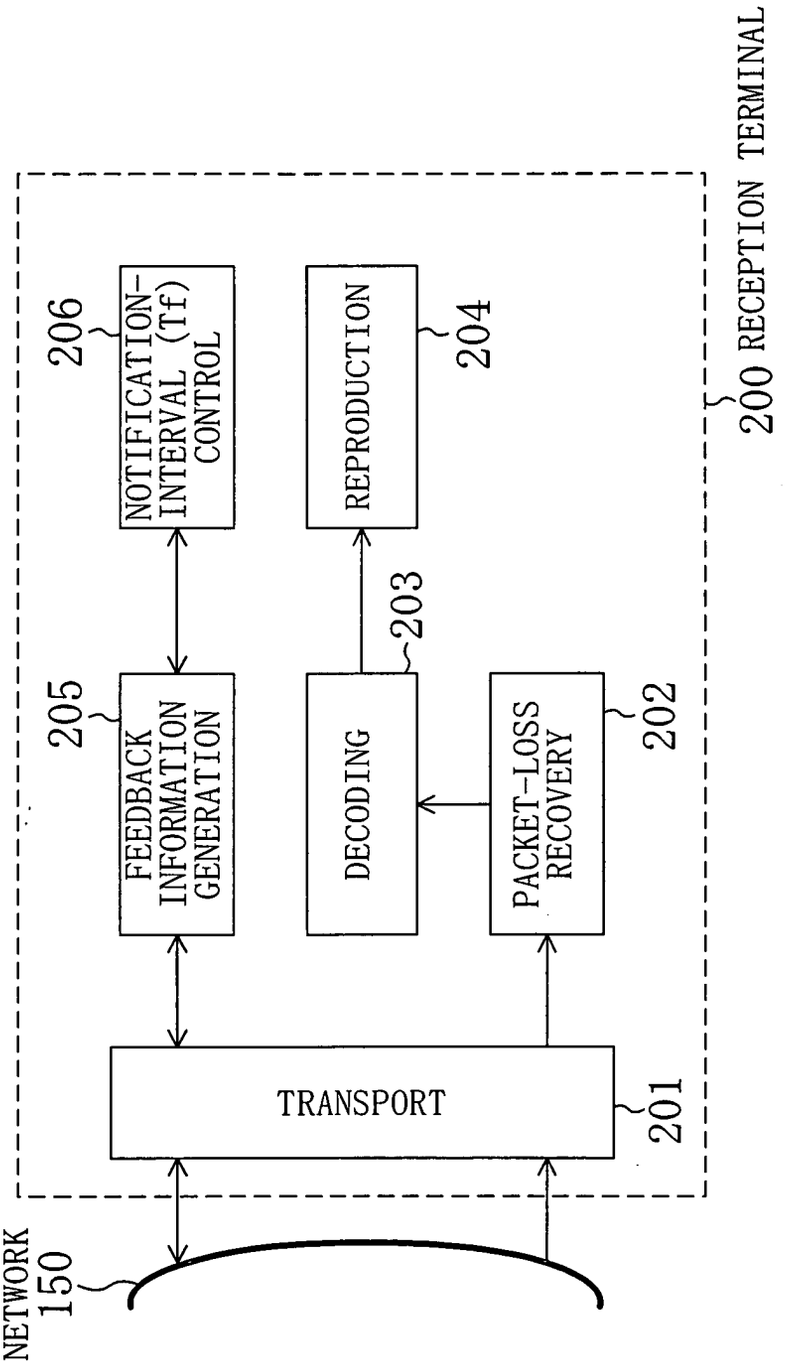

NETWORK
150
TRANSPORT
201
PACKET-LOSS RECOVERY
202
DECODING
203
REPRODUCTION
204
FEEDBACK INFORMATION GENERATION
205
NOTIFICATION-INTERVAL (Tf) CONTROL
206
200 RECEPTION TERMINAL

FIG. 3

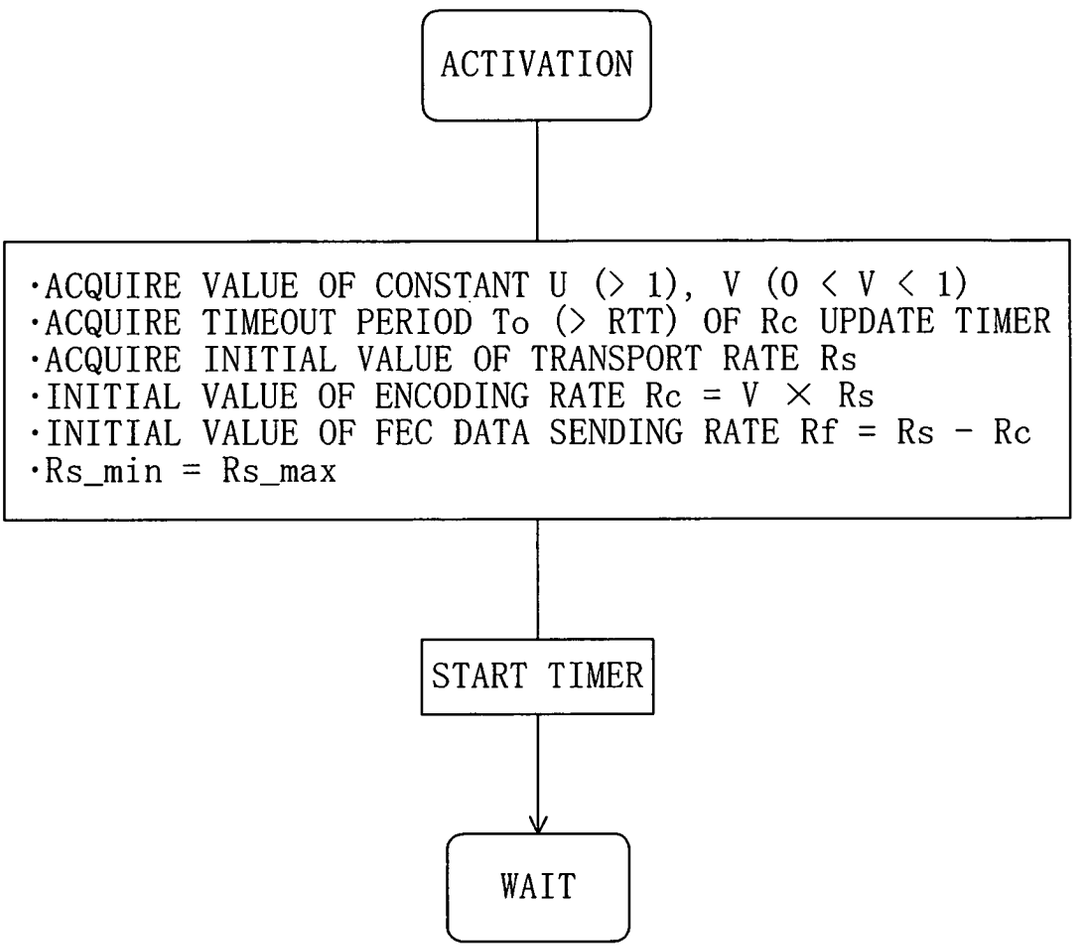

ACTIVATION
·ACQUIRE VALUE OF CONSTANT U (> 1), V (0 < V < 1)
·ACQUIRE TIMEOUT PERIOD To (> RTT) OF Rc UPDATE TIMER
·ACQUIRE INITIAL VALUE OF TRANSPORT RATE Rs
·INITIAL VALUE OF ENCODING RATE Rc = V × Rs
·INITIAL VALUE OF FEC DATA SENDING RATE Rf = Rs - Rc
·Rs_min = Rs_max
START TIMER
WAIT

FIG. 4

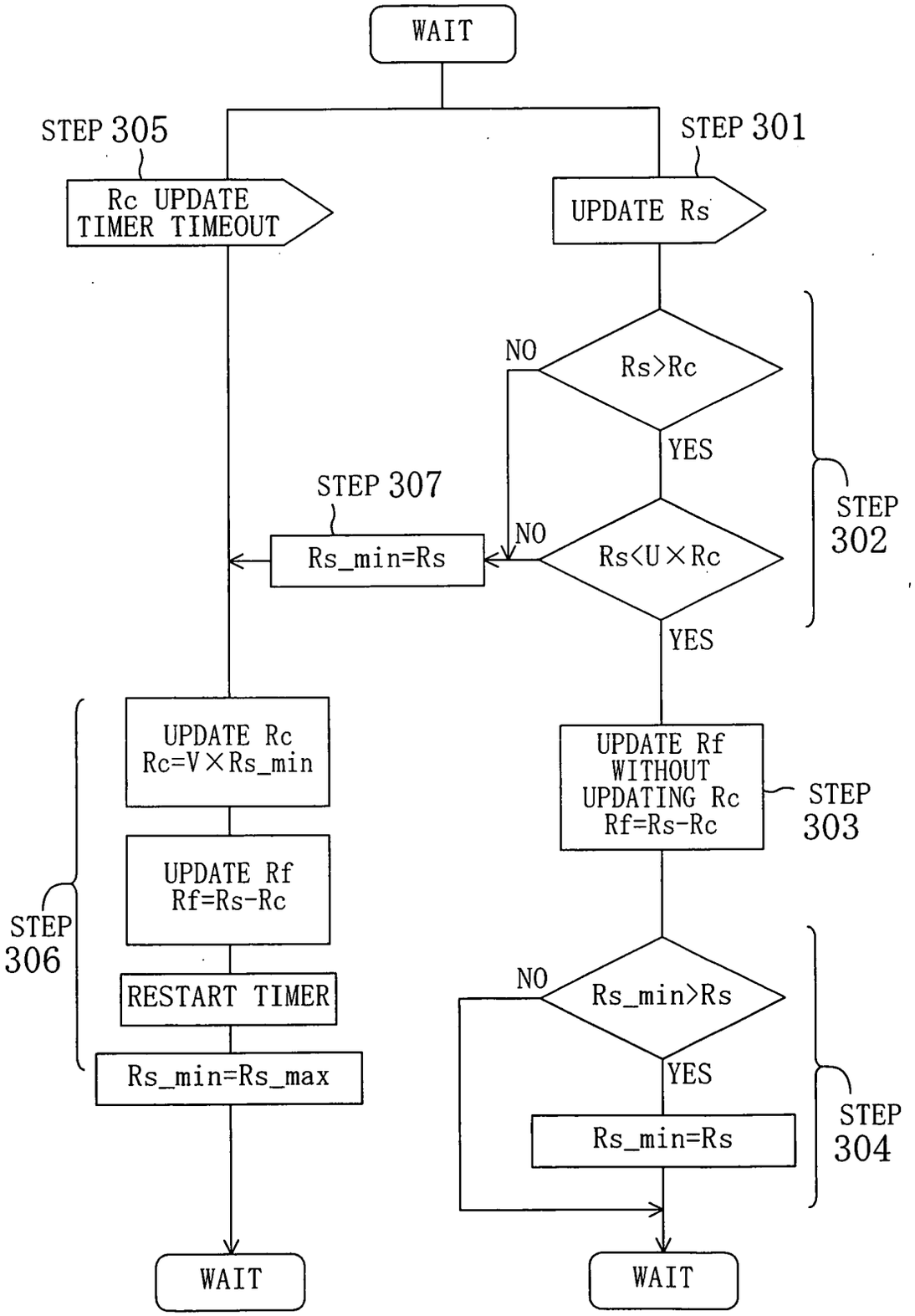

WAIT
STEP 305
Rc UPDATE TIMER TIMEOUT
STEP 301
UPDATE Rs
NO
Rs>Rc
YES
STEP 307
Rs_min=Rs
NO
Rs<U×Rc
STEP 302
YES
UPDATE Rc Rc=V×Rs_min
UPDATE Rf WITHOUT UPDATING Rc Rf=Rs-Rc
STEP 303
UPDATE Rf Rf=Rs-Rc
STEP 306
RESTART TIMER
Rs_min=Rs_max
NO
Rs_min>Rs
YES
Rs_min=Rs
STEP 304
WAIT
WAIT

FIG. 5

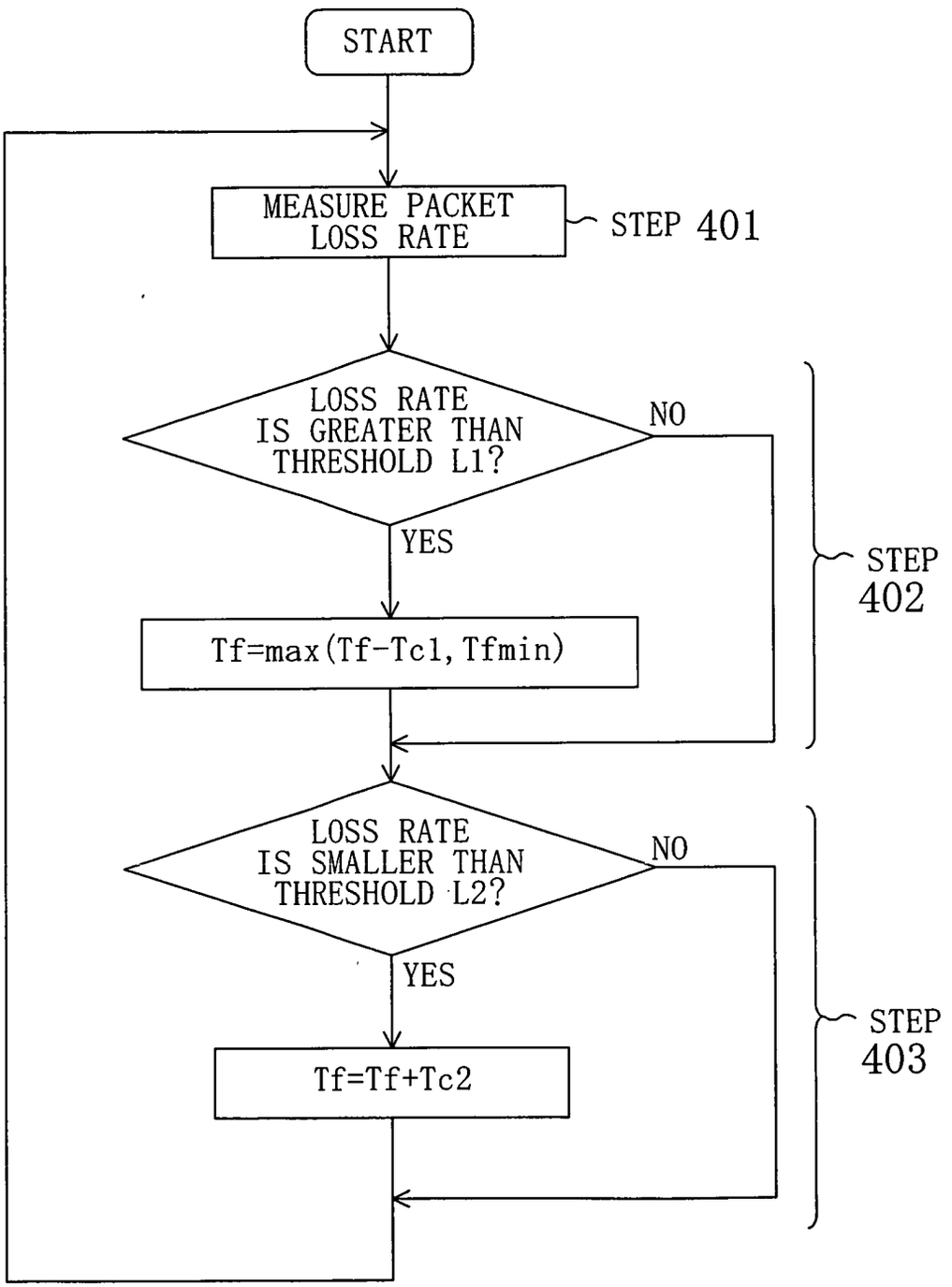

START
MEASURE PACKET LOSS RATE
STEP 401
LOSS RATE IS GREATER THAN THRESHOLD L1?
NO
YES
Tf=max(Tf-Tc1, Tfmin)
STEP 402
LOSS RATE IS SMALLER THAN THRESHOLD L2?
NO
YES
Tf=Tf+Tc2
STEP 403

FIG. 6

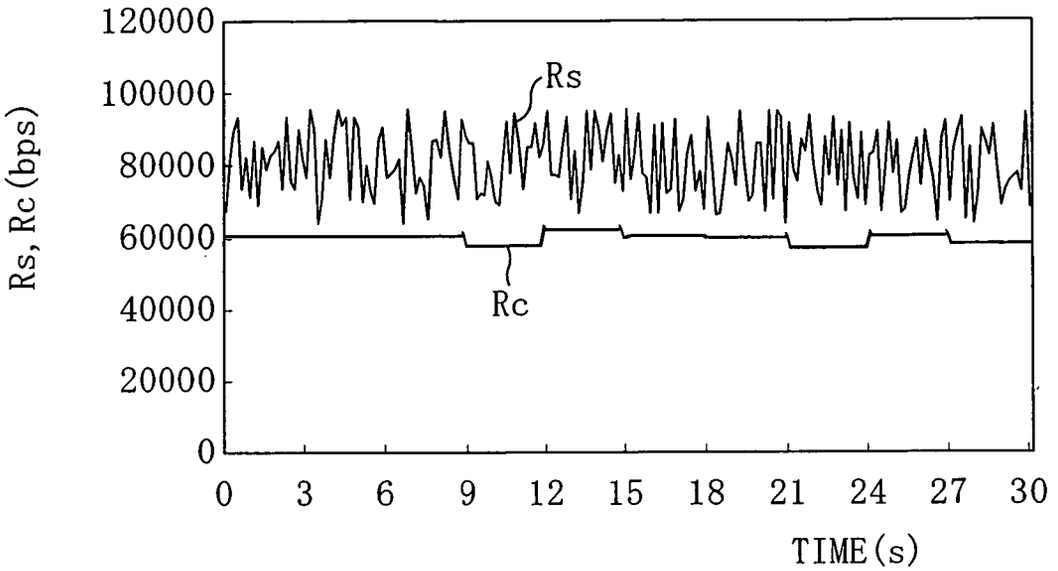

120000
100000
80000
60000
40000
20000
0
Rs, Rc (bps)
Rs
Rc
0
3
6
9
12
15
18
21
24
27
30
TIME(s)

FIG. 9

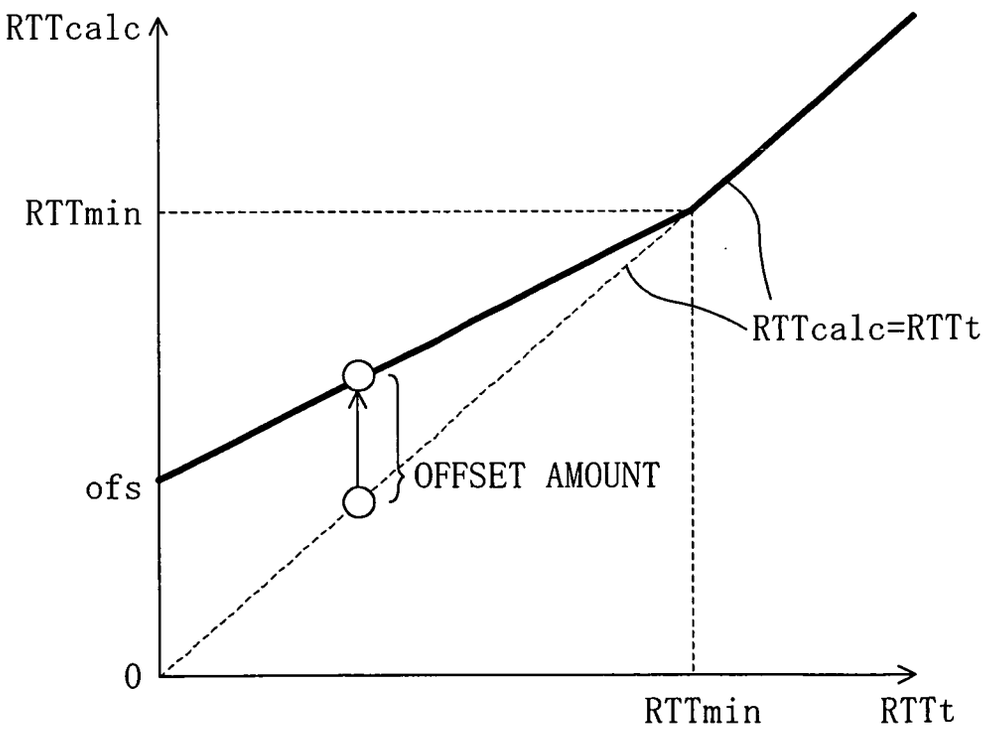

RTTcalc
RTTmin
ofs
0
OFFSET AMOUNT
RTTcalc=RTTt
RTTmin
RTTt

FIG. 7

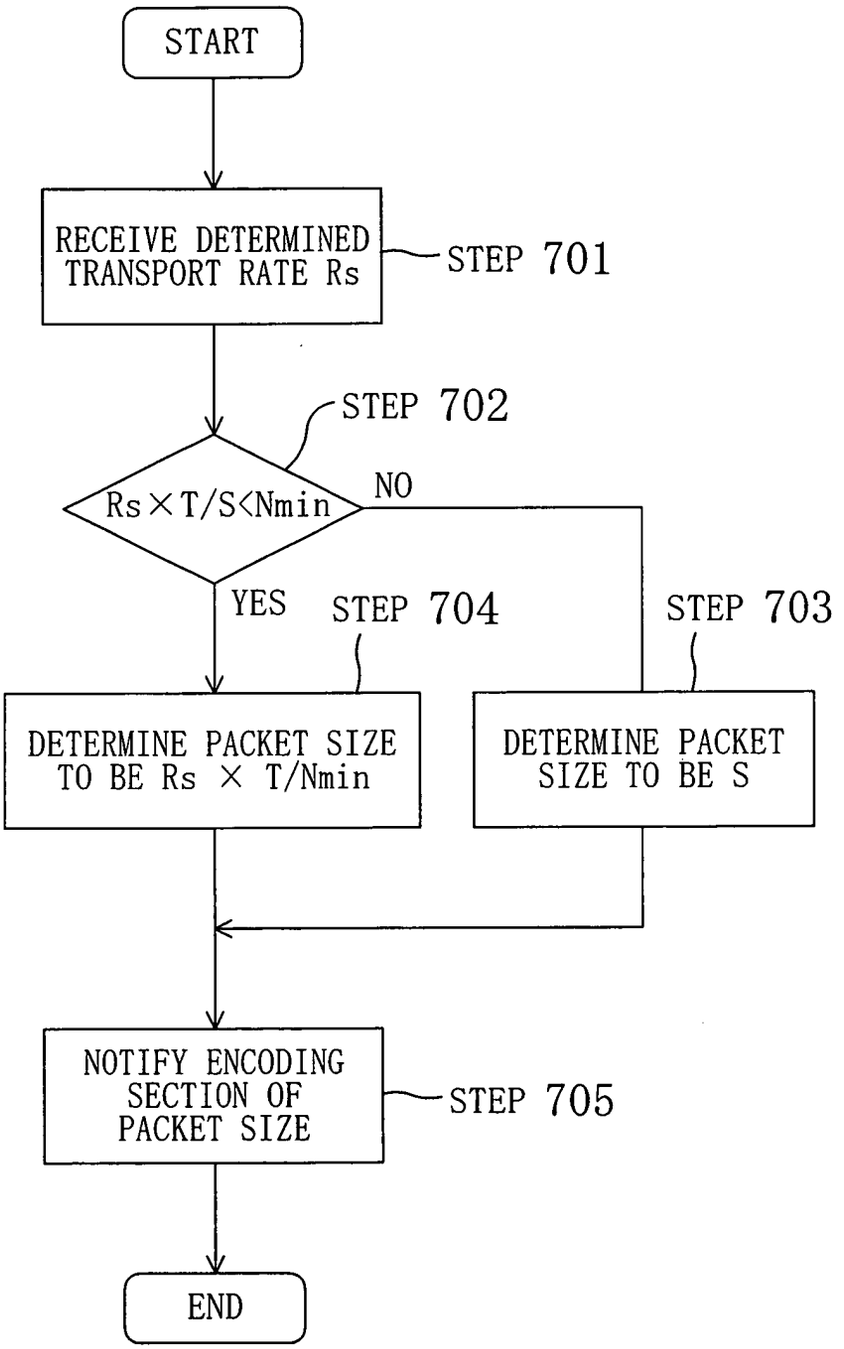
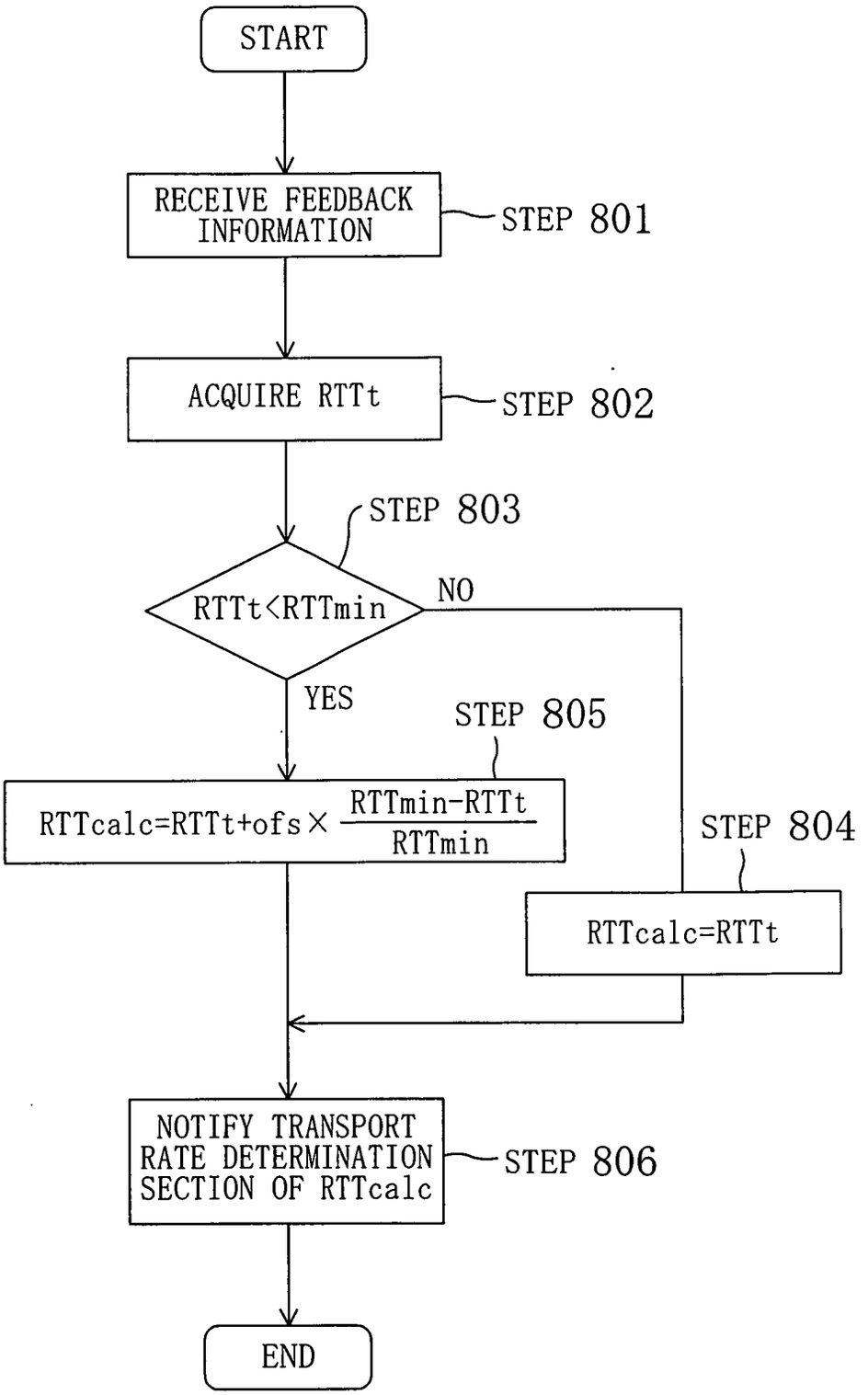

START
RECEIVE DETERMINED TRANSPORT RATE Rs
STEP 701
STEP 702
Rs×T/S<Nmin
NO
YES
STEP 704
STEP 703
DETERMINE PACKET SIZE TO BE Rs × T/Nmin
DETERMINE PACKET SIZE TO BE S
NOTIFY ENCODING SECTION OF PACKET SIZE
STEP 705
END

FIG. 8


START
RECEIVE FEEDBACK INFORMATION
STEP 801
ACQUIRE RTTt
STEP 802
STEP 803
RTTt<RTTmin
NO
YES
STEP 805
RTTcalc=RTTt+ofs×(RTTmin-RTTt)/RTTmin
STEP 804
RTTcalc=RTTt
NOTIFY TRANSPORT RATE DETERMINATION SECTION OF RTTcalc
STEP 806
END